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KAY WARD
TEAM LEADER EXAMINATION
SUPPORT AND SALES

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Patents Act 1990

ORIGINAL

PROVISIONAL SPECIFICATION

MULTIMEDIA NETWORKS

The invention is described in the following statement:

MULTIMEDIA NETWORKS

FIELD OF INVENTION

The present invention relates to the interworking of networks, particularly to the interworking of information through telecommunication networks. In one form, the present invention relates to the interworking of multimedia networks and telecommunication networks based on SS7.

BACKGROUND OF INVENTION

Figure 1 illustrates a Third generation Public Land Mobile Networks (PLMNs). These are capable of operating with compressed voice streams in their core networks. This is voice stream coding which is asynchronous or synchronous in nature, and transmitted and / or received at bit rates in the range typically of 8 - 16 kbit/s.

PSTN/ISDN networks usually operate with synchronous 64kbit/s PCM coded voice streams, which occupies a 64kbit/s time slot. If a call originating in a PLMN transverses a PSTN/ISDN network, in other words, the voice stream passes from a PLMN network to a PSTN/ISDN network, to a terminating PLMN, the voice stream is transcoded from its compressed format to a synchronous PCM coded format and then again transcoded back to its compressed format. This is because there is no ability to negotiate or signal the use of another codec type or bandwidth through the SS7 based network. This is exemplified in Figure 1, where the PSTN/ISDN network is unable to allow an indication of GSM AMR and thus a default PCM coding is required. Transcoding is then necessary. Flowing from this, any gains in bandwidth efficiency on the mobile side are lost because the use of transcoding between codec formats reduces the quality of the original user data. Transcoding should thus be avoided where possible.

By way of background, traditional Telecommunications networks based on SS7 have no such ability to negotiate capabilities between endpoints or network nodes. User plane data is transported through the core network through a 'pipe' with a bandwidth of 64kbit/s multiples based on PCM coding.

With the introduction of Mobile telecommunications networks compressed voice was brought to the edge of the 'fixed' telecommunications networks. The user plane data in a mobile network is encoded by codecs producing

compressed voice. For example: GSM AMR and EFR coding. As a result, a smaller 'pipe' supporting less bandwidth is needed.

The result is that, a network given the same amount of bandwidth can handle more calls.

However, when a fixed telecommunications network interfaces with a mobile network, the compressed voice must be transcoded from the compressed format to the PCM format of the fixed network.

Although, multimedia networks such as those based on H.323, ITU-T H.323, Packet Based Multimedia Communication Systems, and SIP, IETF RFC 2543 Session Initiation Protocol, have the ability to negotiate capabilities on an end to end basis, this cannot be put into effect when the multimedia network is interconnected with one or more SS7 networks.

H.323 uses a protocol called H.245, ITU-T H.245, Control Protocol for Multimedia Communication, which uses OpenLogicalChannel and / or TerminalCapability messages / structures to exchange information between H.323 endpoints about the capabilities supported by the end points.

SIP (Session Initiation Protocol) is defined by the Internet Engineering Task Force (IETF). Part of the SIP multimedia architecture is the Session description protocol (SDP), IETF RFC 2327 Session Description Protocol. It is used to negotiate a set of capabilities between SIP endpoints (SIP server, SIP client etc).

The following is an example of the information that is able to be negotiated by multimedia negotiation protocols:

- Audio (define type and direction - send, receive, send & receive)
- Video (define type and direction - send, receive, send & receive)
- Data (define type and direction - send, receive, send & receive)
- Encryption
- Conference
- Security

Once this negotiation has taken place the user data is transported between the endpoints in the format specified by this negotiation. This negotiation can take place at any time. For example, call setup, in an

established session etc.

Nonetheless, there is still a problem as yet unresolved of when Multimedia networks (e.g. H.323 and SIP) are interfaced with a Telecommunications networks based on SS7 in that they experience the same problem as that associated with mobile networks separated by a PCM network as described above and in that they cannot negotiate capabilities on an end-to-end basis.

For example, the information signalled in the multimedia compatibility protocols such as H.323 and SDP is lost because the SS7 protocols cannot carry this information. As such a node / server / gateway / media gateway is forced to transcode the user data leading to a degradation in quality.

An object of the present invention is to alleviate at least one disadvantage of the prior art.

SUMMARY OF INVENTION

The present invention provides a method of enabling information to pass between a first network based on a first technology and a second network based on a second technology, by enabling the first and / or second networks to apply a method of inband capability negotiation or signalling in order to establish the format for the passing of the information from the first network to the second network.

Preferably, the inband capability is TFO (Tandem Free Operation).

Preferably, the format includes signalling and / or negotiation of protocol and / or of capabilities.

Preferably, the information is further passed from the second network to a third network based on the first technology or a third technology, and in which the step of applying inband capability is effected to the information passing from the second to the third network.

Preferably, the information is user plane information, such as compressed audio stream.

The present invention also provides a communication system having a first network based on a first technology, a second network based on a second technology, and interface means enabling the first and second networks to

apply inband methodology in order to signal and / or negotiate the passing of the information from the first network to the second network.

Preferably, the inband methodology is TFO (Tandem Free Operation).

Preferably, the system further includes a third network based on the first technology or a third technology, and interface means applying TFO to the information transferred from the second to the third network.

Preferably, the networks may be one or more of PLMN network, multimedia network or SS7 based network.

In essence, the present invention realises that inband capability negotiation and / or signalling, such as a protocol called Tandem Free Operation (TFO), ETSI GSM 08.62 Tandem Free Operation, can be introduced to alleviate the problem when interfacing SS7 networks with mobile networks. It allows network nodes to signal and negotiate the type of codec and thus the bandwidth through the network through in band communication. The present invention also alleviates the need for transcoding which would lead to the degradation of quality of the original user plane data.

As TFO provides a means of inband capability negotiation when interfacing with mobile networks, the present invention, in one preferred form, seeks to extend the use of TFO to the interface between multimedia networks (i.e. H.323, SIP) and Telecommunications based on SS7.

Throughout this specification, by 'information' we mean data, voice, video and / or any other form of information electronically transmissible. A preferred form of the invention relates to the transmission of audio or voice streams.

In this specification, a multimedia network is a network that through the use of Call / Session control and / or capability negotiation allows for the control and transport of different types of media, including but not limited to, audio, video, data or different types of transport in data, fixed and mobile. For example H.323, H.324M, SIP based multimedia networks.

DESCRIPTION OF INVENTION

A preferred embodiment of the present invention will now be described with reference to the accompanying drawings, in which:

Figure 1 illustrates a prior art configuration, and

Figure 2 illustrates one example of the present invention.

The present invention stems from problems associated with moving information from one network, of one technology type to another network of another technology type.

Referring to Figure 1, illustrating a prior art configuration, user plane information, generally denoted by the 'line' 1 is passed from an originating network 2, such as PLMN, through a PSTN/ISDN network 3. In order to effect this, the information which in the illustration is compressed voice in network 2 is transcoded at junction 6 by a transcoder(s) 5 to 64 kbit/s for the purposes of this example suitable for network 3. This is because there is no ability to negotiate or signal the use of another codec type or bandwidth through a SS7 based network such as network 3.

If the information is then again passed onto another network 4, the transcoding process is again undertaken at junction 7 by transcoder(s) 5 from 64 kbit/s suitable for network 3 to compressed voice suitable for network 4. Each transcoding process is considered undesirable, and thus there is considered to be advantages in limiting the number of transcoding processes undertaken in an overall network configuration as illustrated in Figure 1.

Referring to figure 2, illustrating a configuration to which the present invention is applicable, again the user plane information is denoted 1, and various networks are denoted 2, 3, and 4. Although the example described refers to multimedia and PSTN/ISDN networks, it is understood that the present invention should not be so limited. Generally, the present invention has application to networks of different technology types, and Figure 2 illustrates only one example of this. In Figure 2, information 1 is passed from network 2 to network 3. However, TFO is utilised in transferring the information, so that ,in the example, a TFO indicating GSM AMR coding is negotiated. There is no transcoding step at the junction of networks 2 and 3.

The result is transcoding is not performed at this point in the overall network, and compressed voice (for this example) is utilised in both network 2 and network 3. Equally, information flowing from network 3 to network 4 is not transcoded and information is passed from network 3 to network 4 by

application of TFO.

In the present invention, transcoding is contemplated at a distal point in the network or a terminal coupled thereto.

TFO is outlined in technical specification as applied to GSM in GSM 08.62 version 7.0.0 Release 1998. The present invention seeks to apply the principles of TFO operation (which allows for negotiation, signalling and use of different GSM audio formats through a telecommunications network using PCM signalling) to the present problem of interfacing between mobile and multimedia networks as outlined above. The specification of TFO only covers the use of audio codecs. To allow the negotiation of video codecs and other capability information, TFO should be extended by applying the TFO principles of negotiating with code points for video codecs and other capability information.

Thus in the example shown for the purposes of illustration, the codec information received in the multimedia capability negotiation protocols (i.e. H.323, SDP) should be mapped to the appropriate values in the TFO protocol and signalled inband and vice versa.

This would allow for the end to end negotiation of capabilities through traditional telecommunication networks based on SS7 and new telecommunications networks based on Transport Independent Call Control (TICC).

Thus the use of transcoders could be reduced resulting in better quality speech and more efficient use of resources in a network.

An example of Mapping between H.245, TFO & SDP follows:

H245 TerminalCapabilitySet

```
TerminalCapabilitySet
{
        sequenceNumber          SequenceNumber,
        protocolIdentifier      OBJECT IDENTIFIER,
```

```
capabilityTable        CapabilityTableEntry
                           CapabilityTableEntryNumber,
                           Capability
                               receiveAndTransmitAudioCapability
                                   gsmEnhancedFullRate
                                       audioUnitSize
                                       comfortNoise
                                       scrambled
    }
```

maps to:

TFO

```
    TFO_REQ_L (Codec GSM EFR)
```

maps to:

SDP

```
    v=0
    o=OwnerSessionID
    s=SessionName
    c=ConnectionInformation
    t=Timesession active
    a=sendrecv
    m=audio PORT Transport GSMEFR
```

In one form, the present invention can reside in an interface between a multimedia network and a traditional telecommunications network using inband signalling.

THE CLAIMS DEFINING THE INVENTION ARE AS FOLLOWS:

1. A method of enabling information to pass between a first network based on a first technology and a second network based on a second technology, the method including the step of:

enabling the first and / or second networks to apply a method of inband capability negotiation or signalling in order to establish the format for the passing of the information from the first network to the second network.

2. A method as claimed in claim 1, wherein the inband capability is TFO (Tandem Free Operation).

3. A method as claimed in claim 1 or 2, in which the format includes signalling and / or negotiation of protocol and / or of capabilities.

4. A method as claimed in claim 1, 2 or 3, wherein the information is further passed from the second network to a third network based on the first technology or a third technology, and in which the step of applying inband capability is effected to the information passing from the second to the third network.

5. A method as claimed in any one of claims 1 to 4, wherein the information is user plane information, such as compressed audio stream.

6. A communication system having

a first network based on a first technology,

a second network based on a second technology, and

interface means enabling the first and second networks to apply inband methodology in order to signal and / or negotiate the passing of the information from the first network to the second network.

7. A system as claimed in claim 6, wherein the inband methodology is TFO (Tandem Free Operation).

8. A system as claimed in claim 6 or 7, further including a third network based on the first technology or a third technology, and interface means applying TFO to the information transferred from the second to the third network.

9. A system as claimed in claim 6, 7 or 8, wherein the third network is one of PLMN or multimedia network.

10. A system as claimed in any one of claims 6 to 9, wherein the second network includes a plurality of telecommunication networks.

11. A system as claimed in any one of claims 6 to 10, in which the information is user plane information, such as a compressed audio stream.

12. A system as claimed in any one of claims 6 to 11, wherein the first network is a multimedia network.

13. A system as claimed in any one of claims 6 to 12, wherein the second network is a SS7 based network.

14. A method as herein disclosed.

15. A system and / or device as herein disclosed.

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ERICSSON AUSTRALIA PTY LTD

WATERMARK PATENT & TRADEMARK ATTORNEYS
290 BURWOOD ROAD
HAWTHORN VICTORIA 3122
AUSTRALIA

RCS/SH DOC 27 AU003178.WPC

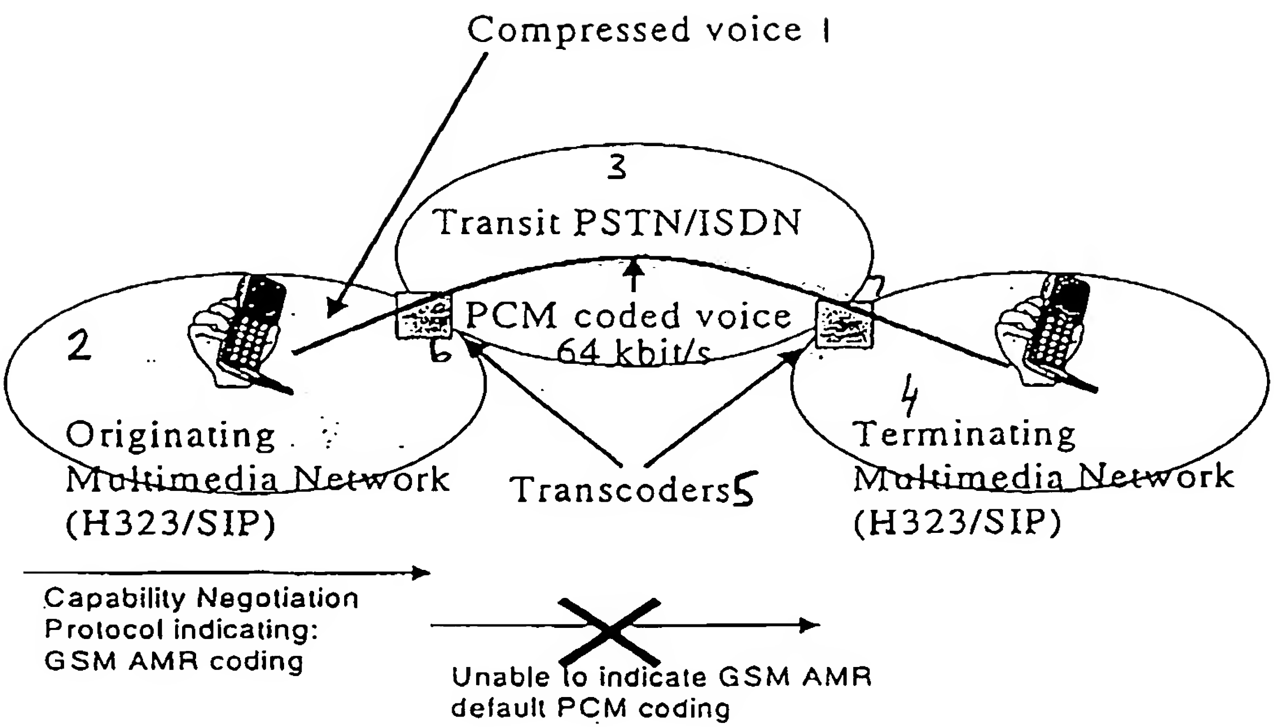
Compressed voice 1
3
Transit PSTN/ISDN
PCM coded voice
64 kbit/s
2
6
4
Originating
Multimedia Network
(H323/SIP)
Transcoders 5
Terminating
Multimedia Network
(H323/SIP)
Capability Negotiation
Protocol indicating:
GSM AMR coding
Unable to indicate GSM AMR
default PCM coding

Figure 1

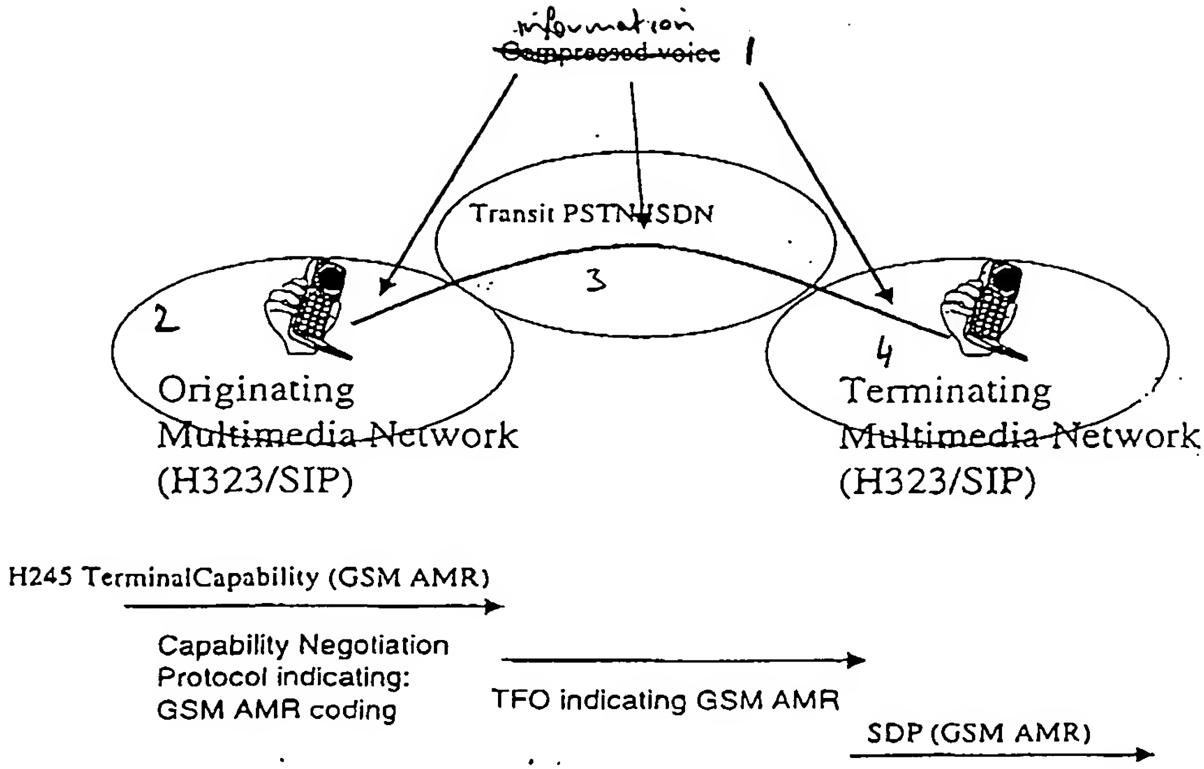
information
~~Compressed voice~~
1
Transit PSTN/ISDN
3
2
Originating
~~Multimedia Ne~~twork
(H323/SIP)
4
Terminating
~~Multimedia N~~etwork
(H323/SIP)
H245 TerminalCapability (GSM AMR)
Capability Negotiation
Protocol indicating:
GSM AMR coding
TFO indicating GSM AMR
SDP (GSM AMR)

Figure 2